DETAILED ACTION

Response to Arguments

Applicant's arguments filed 19 March 2008 have been fully considered but they are not persuasive.

Applicant alleges:

> In response, it is respectfully submitted that even if the adjustable features of Melanson were applied to the hearing aid of Ishige, such a hearing aid would not provide all of the features of claim 1. Specifically, claim 1 recites <u>receiving a user adjustable digital loudness normalization control signal from a user during operation for controlling the configuration of an input/output characteristic having an amount of compression for loudness normalization</u>. Claim 1 further recites that <u>the control signal is configured to be directly increased or decreased by the user during operation for increasing or decreasing the amount of compression</u>. It is submitted that Ishige and Melanson, either alone or in combination, do not provide such a feature.

Applicant alleges that the above underlined limitations are not taught because:

> In particular, with respect to Melanson, the hearing aid allows a user to select between rehabilitation strategies by manipulating a selector switch. The rehabilitation strategies may implement different filter bank structures and compression strategies. Each rehabilitation strategy is implemented as a digital signal processing program on a programmable digital signal processor. However, while each of the rehabilitation strategies may implement different compression strategies, there is no obvious or clear relationship between the amount of compression implemented in each of the rehabilitation strategies and the switch selection.
>
> Specifically, with Melanson's device, the user may select between different programs, which may implement different compression strategies. However, Melanson does not explicitly teach that the different compression strategies have different amounts of compression.

Examiner respectfully disagrees. The claim interpretations can be met byf each one of Melanson's program's changing, and thus changing the compression. This interpretation is not precluded by the claim's current presentation. The user' changing the program is the control signal. This control is received during operation, and effects the compression as shown on page 4 of the outstanding action. While this may not provide an infinite number of intermediate positions as set forth by application (i.e. Nor could they achieve a certain amount of compression in between teh three compression stratetgies; p19 Remarks 3/19/08), the claim is not limited to this feature. Applicant even acknowledges taht one compression strategy may increase the amount of compression, one may decrease and one may not change the amount of compression at all (p.10 Remarks 3/19/08). Thus, when a program is changed,(i.e. a user input 'directly' changes the program) this program will change the compression, directly increasing it or decreasing it.

Applicant further states:

> In addition, for argument's sake, even. if Melanson's device provides different compression strategies for each particular hearing aid program, these compression strategies are combined with a particular hearing aid program and each hearing aid program is designed to provide optimal results in a particular listening environment (see Melanson's abstract) and so the user would be restricted to the compression strategy used for the hearing aid program that corresponds to the user's current listening environment. Furthermore, even if the user found the compression strategy for another hearing aid program to be more beneficial, by selecting that other hearing aid program some parameter values may change based on the listening environment associated with the other hearing aid program which may have an overall detrimental effect on the performance of the hearing aid in the user's current listening environment. The Applicant submits that in both of these situations

> Melanson's device would not be helpful and the Applicant further notes that the Applicant's claimed device would not suffer from these limitations.

Examiner respectfully disagrees. While the user may be restricted to the compression strategy used for the programs that correspond to the listening environments, this interpretation is not precluded by the claims current presentation. All that needs to be present in the prior art is a direct input which will change (increase/decrease) the compression characteristics of the device. Melanson discloses exactly that. While it may not be as flexible as applicant's invention, the claim is not limited to these specific features in its current presentation.

Applicant further alleges:

> With respect to claim 13 of the subject application, the Applicant respectfully submits that the Examiner has failed to respond to the Applicant's comments put forth in the previous response. Specifically, claim 13 recites a plurality of user control signals in which each user control signal is similar to the user control signal claimed in claim 1. The multi-user control allows the user to adjust the control signals separately for each channel in a multi-channel amplification device. The Applicant submits that such a multi-control approach is more flexible than the selector switch taught by Melanson. Whereas in Melanson the user is limited to the small number of programs preset into the instrument, the claimed multi-control of the subject application allows the user to more flexibly set the compression, and hence to affect the gain, of each channel independently of the others and therefore allows for a much wider range or number of combinations of adjustment. The Applicant submits that Melanson does not teach the use of more than one control signal that can be adjusted by the user. The Applicant respectfully requests that the Examiner respond to the Applicant's arguments with respect to claim 13 or withdraw the rejection with respect to claim 13.

Examiner respectfully disagrees. The arguments regarding claim 13 were moot in the last rejection due to the new rejection presented. Regarding Applicant's

concerns, Applicant submits that Melason does not teach the use of more than one control signal that can be adjusted by the user. However, Examiner respectfully disagrees. The claim is not limited to this specific interpretations presented by Applicant. Melanson discloses that any number of environmental settings may be adjusted within the disclosed device. Changing this numerous times would correspond to 'N' user adjustable control signals. Each one of these signals would correspond to specific frequency range adjustments as well as loudness and compression control.

Applicant further states:

> In response, Applicant submits that Ishige and Topholm, neither alone nor in combination, teach all of the features of claim 58. Specifically, neither Ishige nor Topholm teach receiving a user adjustable digital loudness normalization control signal from a user during operation for controlling the configuration of said input/output characteristic having an amount of compression for loudness normalization, the control signal being configured to provide the user with a continually variable control for increasing or decreasing the amount of compression during operation, as recited in claim 58.

Applicant alleges this is not taught because in Topholm, the sliders 12 are not user adjustable. Essentially, “these adjustable sliders are normally covered and not accessible to the user.”

However, in response to applicant's arguments against the references individually, one cannot show nonobviousness by attacking references individually where the rejections are based on combinations of references. See *In re Keller*, 642 F.2d 413, 208 USPQ 871 (CCPA 1981); *In re Merck & Co.,* 800 F.2d 1091, 231 USPQ 375 (Fed. Cir. 1986).

The rejection was made under 35 U.S.C. 103(a), an obviousness rejection. While these sliders are not "normally" available to the user, it does not mean that they are never, and that it would not be obvious to provide them to a user. In the combination of the rejection, the sliders are added to Ishige to allow for adjustment. The combination is a new device, obvious to one of ordinary skill in the art, Separate from the restrictions and limitations of both Ishige and Topholm. Topholm provides the means for performing the claimed limitations, but does not necessarily anticipate them in the manner they are claimed. However, its obvious that these means can be adapted for use by any particular person at any particular time. That is the case in the outstanding rejection.

Furthermore, these means can be adjusted at any time. While they may be required to be stored (as stated by Applicant) the fact that a user of the combination can adjust them at any time, implies that they are dynamic in nature.

Claim Rejections - 35 USC § 103

The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

Claim 1 – 40, 42, 43 and 45 - 57 are rejected under 35 U.S.C. 103(a) as being unpatentable over Ishige (U.S. Patent 5,892,836) in view of Melanson (U.S. Patent 6,104,822).

Regarding **Claims 1, 13 and 33**, Ishige discloses:

A method of generating an analog acoustic output signal from an acoustic input signal in accordance with a configurable input/output characteristic (abstract), said method comprising the steps of:

(a) converting the acoustic input signal into a digital acoustic input signal (i.e. and input circuit for receiving the analog audio signal and converting it into a digital signal; Fig. 7 element 12);

(b) transforming the digital acoustic input signal into one or more frequency domain input signals (Fig. 6 and col. 7 lines 37 – 57);

(c) detecting the magnitude of each of the one or more frequency domain input signals (i.e. a frequency analyzer; Fig. 7 element 21).

Ishige does not explicitly disclose:

(d) receiving a user adjustable digital loudness normalization control signal from a user during operation for controlling the configuration of said input/output characteristic having an amount of compression for loudness normalization, the control signal being configured to be directly increased or decreased by the user for increasing or decreasing the amount of compression during operation

(e) for each of the one or more frequency domain input signals, determining a gain value in response to the user adjustable digital loudness normalization control signal and the magnitude of the frequency domain input signal.

(f) providing one or more frequency domain output signals by multiplying each of the frequency domain input signals by the corresponding gain value.

(g) transforming the one or more frequency domain output signals into a digital acoustic output signal:

(h) converting the digital acoustic output signal into the analog acoustic output signal.

Melanson discloses a digital signal processor hearing aid with a program selector switch (Fig. 1a element 46) that is preferably manipulable by a user to allow the user to dynamically select which of the digital signal processing means to invoke in which listening environment. In dealing with these environments, each of the processing means may implement such functions as *compression*, noise compensation, feedback cancellation, etc; col. 8 lines 30 – 50.

Applying this environmental selection switch to Ishige would allow the user to conveniently alter the characteristic's of Ishige's hearing aid to further assist the user in various environments.

Modifying Ishige to include the selection feature taught by Melanson discloses:

(d) receiving a user adjustable digital loudness normalization control signal from a user during operation for controlling the configuration of said input/output characteristic having an amount of compression for loudness normalization, the control signal being configured to be directly increased or decreased by the user for increasing or decreasing the amount of compression during operation (col. 8 lines 30 – 50; col. 19 lines 10 – 15 of Melanson; the switch of Melanson alters the characteristics such as compression, since it is set for various environments, changing it will either directly increase it or decrease it accordingly to a set level);

(e) for each of the one or more frequency domain input signals, determining a gain value in response to the user adjustable digital loudness normalization control signal and the magnitude of the frequency domain input signal (i.e. the hearing compensating filter coefficient circuit receives the analysis result and the hearing characteristics of the person and sets the filer coefficients; col. 7 lines 57 – 67 and col. 8 lines 1 – 10; this portion of Ishige functions as the various DSPs in Fig. 1a of Melanson);

(f) providing one or more frequency domain output signals by multiplying each of the frequency domain input signals by the corresponding gain value (i.e. the digital audio signal is supplied to the hearing compensating circuit; Fig. 7 element 22);

(g) transforming the one or more frequency domain output signals into a digital acoustic output signal (i.e. the output of the hearing compensation circuit is applied to the output circuit; Fig. 7);

(h) converting the digital acoustic output signal into the analog acoustic output signal (Fig. 7 element 13).

It would have been obvious to one of ordinary skill at the time of the invention to apply the adjustable features of Melanson to the hearing aide of Ishige. Melanson discloses that previous hearing aids are typically capable of only providing one single strategy with adjustable parameters. This is similar to what Ishige discloses. Melanson further states that in the hearing aid of Fig. 1a, the hearing aid can implement more than one strategy and thus is better able to adapt and to provide optimal results in a variety of different listening environments.

Regarding **Claims 2, 14, 34 – 37 and 57**, in addition to the elements stated above regarding claims 1, 13, and 33, the combination further discloses:

wherein step (d) further comprises adjusting the configurable input/output characteristic for at least one frequency band corresponding to the one or more frequency domain input signals by one of:

increasing the level of said configurable input/output characteristic by a larger amount for lower level sounds compared to higher level sounds when a user adjusts the user adjustable digital loudness normalization control signal to increase the level of the analog acoustic output signal,

and decreasing the level of said configurable input./output characteristic by a smaller amount for lower level sounds compared to higher level sounds when the user adjusts the user adjustable digital loudness normalization control signal to decrease the level of analog acoustic output signal (i.e. in the combination, Ishige's hearing aid is adjustable as taught by Melanson. Melanson discloses that the invention is adjustable to different listening environments thus optimizing the amplification to fit the environment; cols 8 and 9. Thus, adjusting the hearing aid of the combination increases and decreases the level of the input/output characteristic dependent upon the setting of the user in various environments).

Regarding **Claims 3, 15 and 50**, in addition to the elements stated above regarding claims 1, 13 and 33, the combination further discloses:

performing steps (c), (e) and (f) by means of a programmable processor (i.e. Ishige discloses (*processor*) elements 21, 22, 24, 25, 26, 27 and 31 in Fig. 7 that process digital signals and are programmed via the memory through the fitting device).

Regarding **Claims 4, 21, 39 and 53**, in addition to the elements stated above regarding claims 3, 15, 34 and 50, the combination further discloses:

wherein step (e) comprises calculating the corresponding gain value for the one or more frequency domain input signals by means of a fitting formula programmed into said programmable digital signal processor, wherein a parameter of the fitting formula is provided by the user adjustable digital loudness normalization control signal (i.e. the hearing compensating filter coefficient setting circuit defines the coefficients for the filters in the hearing compensating circuit on the data that is in the memory which is supplied by the user (Fig. 7 and the associated text in the disclosure) through the input of Melanson)

Regarding **Claims 5, 22, 38 and 51**, in addition to the elements stated above regarding claims 3, 15, 34 and 50, the combination further discloses:

wherein step (e) comprises determining the corresponding gain value for each of the one or more frequency domain input signals by means of a look-up table stored in said programmable digital signal processor, wherein information in the look-up table is retrieved based on the user adjustable digital loudness normalization control signal (i.e. the coefficient table in Fig. 7; and at the time of changing the characteristics of the hearing compensating filter, the channel filter coefficient setting circuit refers to the

coefficients stored in the coefficient table; col. 8 lines 45 – 55; and the hearing compensating circuit is configured to cause the input audio signal to match with the narrowed dynamic range of the person fitted with the hearing aid; col. 7 lines 4 – 7; which is adjustable as taught by Melanson above).

Regarding **Claims 6, 23, 24 and 52**, in addition to the elements stated above regarding claims 5, 22 and 50, the combination further discloses wherein said look-up table is stored in non-volatile memory in said programmable digital signal processor (i.e. Ishige discloses (*processor*) elements 21, 22, 24, 25, 26, 27 and 31 in Fig. 7 that process digital signals (*a digital signal processor*) and are programmed via the memory through the fitting device; and the coefficient table can be stored in ROM; col. 8 line 55).

Regarding **Claims 7, 25, 40 and 54**, in addition to the elements stated above regarding claims 3, 15, 34 and 50, the combination further discloses:

wherein step (e) comprises determining the corresponding gain value for each of the one or more frequency domain input signals by means of a fitting formula programmed into said programmable digital signal processor and a look-up table, wherein information in the look-up table is retrieved based on the user adjustable digital loudness normalization control signal (i.e. the coefficient table in Fig. 7; and at the time of changing the characteristics of the hearing compensating filter, the channel filter coefficient setting circuit refers to the coefficients stored in the coefficient table; col. 8 lines 45 – 55; and the hearing compensating circuit is configured to cause the input

audio signal to match with the narrowed dynamic range of the person fitted with the hearing aid; col. 7 lines 4 – 7; which is adjustable as taught by Melanson above).

Regarding **Claims 8, 16, 17, 26, 27 and 55**, in addition to the elements stated above regarding claims 7, 5, 25 and 50, the combination further discloses wherein said look-up table is stored in non-volatile memory in said programmable digital signal processor (i.e. Ishige discloses (*processor*) elements 21, 22, 24, 25, 26, 27 and 31 in Fig. 7 that process digital signals (*a digital signal processor*) and are programmed via the memory through the fitting device; and the coefficient table can be stored in ROM; col. 8 line 55).

Regarding **Claim 9**, in addition to the elements stated above regarding claim 1, the combination further discloses:

wherein step (b) comprises transforming the digital acoustic signal into at least two frequency domain input signals, each of said frequency domain input signals having a configurable channel input/output characteristic associated therewith, said configurable channel input/output characteristic together forming said configurable input/output characteristic, and wherein said at least two frequency domain input signals are provided with different channel input/output characteristics (i.e. Fig. 6 and col. 7 lines 37 – 57; and the hearing compensating filter coefficient circuit receives the analysis result and the hearing characteristics of the person and sets the filer coefficients; col. 7 lines 57 – 67 and col. 8 lines 1 – 10).

Regarding **Claims 10 – 12, 20, 28 – 30 and 45 - 49**, in addition to the elements stated above regarding claims 1, 13 and 34, the combination further discloses wherein said configurable input/output characteristic is a curvilinear compression characteristic, an input compression characteristic, and an output compression characteristic. Ishige discloses matching the input audio signal with the narrowed dynamic range of the person fitted with the hearing aid using a filter; col. 7 lines 4 – 7. Thus, depending on the users hearing loss characteristics, the device may increase or decrease the high and low frequency components at different values. Further portions will lower amplitude values may be increased or decreased accordingly. As such, Ishige anticipates this element of the claimed invention. Additionally, the device is adjustable as taught by Melanson above.

Regarding **Claims 18, 19, 31 and 32**, in addition to the elements stated above regarding claims 9, 2, 13 and 33, the combination further discloses:

wherein each of said configurable channel input/output characteristics are (is) varied in response to said user adjustable digital loudness normalization control signal (i.e. the memory is programmed/fitted to the hearing characteristic of the user; and the hearing compensating circuit is configured to cause the input audio signal to match with the narrowed dynamic range of the person fitted with the hearing aid; col. 7 lines 4 – 7).

Regarding **Claim 56**, in addition to the elements stated above regarding claim 33, the combination further discloses:

(a) a microphone for receiving an input sound providing an analog input acoustic signal (Fig. 7 element 11)

(b) an A/D converter coupled to said sound energy signal and for reception device for receiving said analog input acoustic signal or an image of said analog input acoustic signal and coupled to said analysis filter for providing said digital acoustic input signal (Fig. 1 element 12)

(c) a D/A converter coupled to said synthesis filter for receiving said digital output acoustic signal and for providing an analog output acoustic signal (Fig. 1 element 13);

(d) a speaker coupled to said D/A converter for receiving said analog output acoustic signal and providing an output sound energy signal (Fig. 1 element 14).

Regarding **Claims 42 and 43**, claims 42 and 43 claim various methods of which to adjust the control signal, a variable resistor and a two-way switch which are not explicitly disclosed by the combination. However, Melanson discloses a user manipulable switch but does not provide details on its implementation.

Examiner takes official notice that various different switches exist including variable resistors (i.e. potentiometers as shown by Martin US 6,104,822 previously) and mechanical switches.

As Melanson is silent it is clear that the implementation of the switch is not a key feature. Many different switches exist and their implementations in the art of hearing aides (such as the one of the combination) do not produce any new or unexpected results. In other words, using a mechanical two-way or a potentiometer would not

patentably distinguish the claimed invention from the prior art because it does not produce any new or unexpected result.

Thus implementing a programming means such as the method disclosed by Martin or a two way switch would have been obvious to one of ordinary skill in the art. One would have been motivated to do so to in order to effectively enter an input in to the combination's hearing aid.

Claim 58 is rejected under 35 U.S.C. 103(a) as being unpatentable over Ishige (U.S. Patent 5,892,836) in view of Topholm (U.S. Patent 4,947,432).

Regarding **Claim 58**, Ishige discloses:

A method of generating an analog acoustic output signal from an acoustic input signal in accordance with a configurable input/output characteristic (abstract), said method comprising the steps of:

(a) converting the acoustic input signal into a digital acoustic input signal (i.e. and input circuit for receiving the analog audio signal and converting it into a digital signal; Fig. 7 element 12);

(b) transforming the digital acoustic input signal into one or more frequency domain input signals (Fig. 6 and col. 7 lines 37 – 57);

(c) detecting the magnitude of each of the one or more frequency domain input signals (i.e. a frequency analyzer; Fig. 7 element 21).

Ishige does not explicitly disclose:

(d) receiving a user adjustable digital loudness normalization control signal from a user during operation for controlling the configuration of said input/output characteristic having an amount of compression for loudness normalization, the control signal being configured to provide the user with a continually variable control for increasing or decreasing the amount of compression during operation.

(e) for each of the one or more frequency domain input signals, determining a gain value in response to the user adjustable digital loudness normalization control signal and the magnitude of the frequency domain input signal.

(f) providing one or more frequency domain output signals by multiplying each of the frequency domain input signals by the corresponding gain value.

(g) transforming the one or more frequency domain output signals into a digital acoustic output signal:

(h) converting the digital acoustic output signal into the analog acoustic output signal.

Topholm discloses a programmable hearing aid with a multifunction sliding switch (Figs. 3 and 5).

Applying these sliding switches to Ishige would allow the user to conveniently alter the characteristic's of Ishige's hearing aid to further assist the user in various environments.

Modifying Ishige to include the selection feature taught by Topholm discloses:

(d) receiving a user adjustable digital loudness normalization control signal from a user during operation for controlling the configuration of said input/output characteristic having an amount of compression for loudness normalization, the control

signal being configured to be directly increased or decreased by the user for increasing or decreasing the amount of compression during operation (Sliders 12 of Topholm added to Ishige which allow for adjustment of such elements as compression, frequency and output power; col.3);

(e) for each of the one or more frequency domain input signals, determining a gain value in response to the user adjustable digital loudness normalization control signal and the magnitude of the frequency domain input signal (i.e. the hearing compensating filter coefficient circuit receives the analysis result and the hearing characteristics of the person and sets the filer coefficients; col. 7 lines 57 – 67 and col. 8 lines 1 – 10 in Ishige);

(f) providing one or more frequency domain output signals by multiplying each of the frequency domain input signals by the corresponding gain value (i.e. the digital audio signal is supplied to the hearing compensating circuit; Fig. 7 element 22 in Ishige);

(g) transforming the one or more frequency domain output signals into a digital acoustic output signal (i.e. the output of the hearing compensation circuit is applied to the output circuit; Fig. 7 in Ishige);

(h) converting the digital acoustic output signal into the analog acoustic output signal (Fig. 7 element 13 in Ishige).

It would have been obvious to one of ordinary skill at the time of the invention to apply the adjustable features of Topholm to the hearing aide of Ishige. One would be motivated to do so to apply a known technique (Topholms adjustable characteristics) to a known device (Ishige's hearing aid and Hearing aids in general) ready for

improvement (i.e. user adjustment) to yield predictable results (i.e. to allow user adjustment of Ishige's hearing aid as is notoriously well known desirable feature in the art).

Conclusion

THIS ACTION IS MADE FINAL. Applicant is reminded of the extension of time policy as set forth in 37 CFR 1.136(a).

A shortened statutory period for reply to this final action is set to expire THREE MONTHS from the mailing date of this action. In the event a first reply is filed within TWO MONTHS of the mailing date of this final action and the advisory action is not mailed until after the end of the THREE-MONTH shortened statutory period, then the shortened statutory period will expire on the date the advisory action is mailed, and any extension fee pursuant to 37 CFR 1.136(a) will be calculated from the mailing date of the advisory action. In no event, however, will the statutory period for reply expire later than SIX MONTHS from the mailing date of this final action.

Any inquiry concerning this communication or earlier communications from the examiner should be directed to ANDREW C. FLANDERS whose telephone number is (571)272-7516. The examiner can normally be reached on M-F 8:30 - 5:00.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Suhan Ni can be reached on (571) 272-7505. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free). If you would like assistance from a USPTO Customer Service Representative or access to the automated information system, call 800-786-9199 (IN USA OR CANADA) or 571-272-1000.

/Andrew C Flanders/
Patent Examiner
Art Unit 2615

/Suhan Ni/
Primary Examiner, Art Unit 2615